subjective impairment test results this only applied when the signal's audio quality was already rated in the "very annoying" subjective range.

With digital audio, the system outage with all systems tested is much more abrupt than in the case of current analog modulation (AM & FM). Therefore, with the introduction of digital audio, broadcasters and their audiences will experience a new characteristic of signal failure at the limit of service.

Other Findings

1. **Signal reacquisition**: Reacquisition times in excess of 1.00 seconds are likely to exceed a maximum threshold of consumer acceptance. The only digital systems that meet that criteria are Eureka-147 and AT&T/Lucent Technologies.

2. **Field Testing**: Field testing confirms the superiority of the Eureka-147/DAB system's robustness with signal impairments. The single transmitter testing showed considerable coverage capability. The multiple transmitter tests confirmed extension of coverage and fill-in capabilities and the benefits of a properly planned multi-transmitter system.

3. Field tests also illustrated the **VOA/JPL satellite system's susceptibility** to extensive signal blockages and the resultant audio failures (or mutes) caused by terrain, buildings, signs, trees and other foliage. This is primarily due to effects of the frequency used for testing (S-band). It has been suggested that these outages may be mitigated to a degree by using a variety of technical means including: high powered satellites, higher elevation angles, spatial and time diversity, and terrestrial repeaters.

4. **IBOC System Modifications.** USA Digital Radio has presented information from other studies of its IBOC system(s) separate from the DAR Subcommittee test program. That information has detailed the critical problems resulting from its current design(s), and in some respects confirmed the findings reflected in these laboratory test results. Reports from USADR suggested new system designs intended to improve performance in some areas. See Appendices 8 & 9. Proposed changes include RF spectral occupancy, power ratios, modulation format, sideband diversity and time diversity as well as digital coding . Also proposed is a **hybrid IBOC** system that defaults to analog during digital system impairments to provide program continuity but at the expense of keeping the analog channel with duplicate programming, time synchronized, and with resulting analog audio (degraded) performance. These latest proposed changes are separate from and in addition to those made earlier by the proponent which precipitated a second round of laboratory testing.

   Appendices 10 & 11 assesses the ability of these system changes to improve the IBOC DAR concept to the point of acceptability. The precise technical parameters have not revealed by USADR, but parameters deduced from the information that has been revealed have been applied in these analyses. There, it is shown that the fundamental design tradeoffs successfully achieving audio quality, compatibility and performance have yet to be achieved.

A more recent proposal (attached as Appendix 12, presented by USADR April, 1997) analyzes much of the information derived from the laboratory testing program, confirms those results, and offers further design considerations.

Future IBOC technology developments producing demonstrably improved performance should be studied further. However, it is unlikely that performance improvement can be such that audio quality, RF compatibility, performance with channel impairments, and extent of coverage problems can all be resolved at the same time.

5. **Eureka-147/DAB**: The Eureka-147/DAB system showed superior performance in all areas of quality and digital signal robustness. As a new-band system, compatibility with other services was not an issue and this advantage is reflected in the system design and performance.

VI. Conclusions

The IBOC systems are not feasible at this time due to deficient performance in the areas studied: audio quality, performance with channel impairments, RF compatibility and extent of coverage.

The IBAC system cannot be deployed due to interference with the current spectrum occupancy of the FM band.

The VOA/JPL system at S-band frequencies is subject to continuous and/or repeated outages due to blockage. It is not clear that this could be totally remedied.

Of all the systems tested, only the Eureka-147/DAB system offers the audio quality and signal robustness performance that listeners would expect from a new DAR service in all reception environments.

Acknowledgement

The tireless efforts of many dedicated engineers and scientists made these evaluations of DAR systems possible, most notably Thomas B. Keller, David Londa, Robert McCutcheon, Stanley Toncich, Gerald Chouinard, Louis Thibault, Ted Grusec, Gilbert Soulodre, Robert Culver, Michael Grimes, Stanley Salek, Daniel Mansergh, and a host of others. A special acknowledgement is due James Hollansworth and the NASA Lewis Research Center for their continued support of and cooperation with EIA/CEMA in its efforts to arrive at sound technical basis for DAR system comparisons.

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Appendices

1. Further System Descriptions

2. Subjective Assessments of Audio Quality of DAR Systems

3. Digital Sound Broadcasting Impairment Test Results

4. Summary of In-Band Terrestrial System Laboratory Test Results

5. Measured analog FM Signal Levels and Impact on In-band Digital Audio Broadcasting Implementation

6. FM Modulation Increasing Baseband Noise In The Presence of An IBOC Digital Signal

7. In-Band Digital Sound Broadcasting Subcarrier Tests

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13. Summary of DAR Field Testing Procedures

Appendix 1

Further System Descriptions

A. AT&T/Lucent Technologies

The AT&T digital audio radio system is designed to operate in the In-band Adjacent Channel (IBAC) mode or In-Band Reserved Channel (IBRC) mode in the 88-108 FM radio band. The digital signal occupies a single 200 kHz FM channel. Digital audio coding is provided by the AT&T Perceptual Audio Coder (PAC) which provides a 160 kbps signal for the laboratory and field tests.

The system uses a 4-phase modem, an adaptive channel equalizer and a three-layer method of error protection to maintain audio quality in the presence of transmission impairments.

The input of the 4-phase modulator is a 360 kbps bit stream (composed of 340 kbps of multiplexed audio data and an overhead of 20 kbps for synchronization and channel equalization). The 4-phase modulation provides an ideal efficiency of 2 bits/sec/Hz, and an actual rate of 1.8 bits/sec/Hz in packing the 360 kbps data into a 200 kHz FM channel. The RF spectrum of this DAR system includes a pilot tone to aid in efficient carrier recovery.

A PN sequence is used for estimating the channel impulse response, and also for bit synchronization.

B. AT&T/Amati/Lucent Technologies

The second digital audio radio system proposed by AT&T is an In-band On-Channel (IBOC). The composite DAR/FM signal in this IBOC system is intended to conform to the FCC PSD masks. Digital audio coding is provided by the AT&T Perceptual Audio Coder (PAC) which provides a 160 kbps digital signal for a stereo audio channel.

The equipment that was delivered to the laboratory for testing operated in three modes, Double Sideband (DSB), Lower Sideband (LSB), or Upper Sideband (USB) modes. The system was tested in the DSB and LSB modes. In the DSB mode the digital signal is located in a 73.5 kHz wide sideband (sidelobe) that runs from 126.5 kHz to 200 kHz above and below the FM channel center frequency. In the LSB and DSB mode one sideband (sidelobe) is used. In the DSB mode the total composite bandwidth is 400 kHz. In the DSB mode the digital signal average power is about 15 dB below the host FM.

The IBOC signal uses discrete multitone or COFDM modulation. The subcarrier spacing is 4 kHz. The symbol duration is 250 microseconds. In the DSB mode 32 subcarriers are used, and in the LSB or USB mode 18 subcarriers are used. Differential 4-phase modulation is used for the DSB mode, and for the LSB or USB modes (sidelobe) 8-phase modulation is used.

C. Eureka 147

The system tested in the laboratory and for the field tests operated in the 1452 to 1492 MHz (L band). This system occupies a bandwidth of 1.5 MHz and is capable of transmitting multiple audio channels. The System uses Coded Orthogonal Frequency Division Multiplex (COFDM) modulation. The number of radiated carriers is 384 for the E-147 mode II. The useful symbol duration is 250 microseconds.

Digital audio coding uses the MUSICAM system at a bit rate of 224 kbps for both field and laboratory tests. A second mode was tested in the laboratory at an audio rate of 192 kbps. The Eureka 147 system tested is capable of transmitting five stereo channels, one coded at 256 kbps, two at 224 kbps, and two at 192 kbps. With the five stereo pairs a mono 64 kbps, 64 kbps data, and 24 kbps data channels may be added.

D. VOA/JPL

This system is designed for direct to the listener satellite distribution. The system is designed to operate in the 2310 to 2360 MHz (S band). Because of satellite transponder availability, the system was field and laboratory tested at 2030 MHz.

The digital signal occupies a single 200 kHz channel. Digital audio coding is provided by the AT&T Perceptual Audio Coder (PAC) which produces a 160 kbps signal for the laboratory tests. For power and bandwidth efficiency, Quadrature Phase Shift Keying (QPSK) modulation with root raised cosine pulse shaping is used.

E. USADR FM-1

The composite DAR/FM signal in this IBOC system is intended to conform to the FCC PSD masks. The FM-1 stereo audio source coding rates vary from a minimum of 128 kbps to a maximum of 256 kbps on a frame-by-frame basis.

The IBOC digital signal is located in a 100 kHz wide sideband (sidelobe) that runs from 120 kHz to 220 kHz above and below the FM channel center frequency for a total composite channel bandwidth (3 dB) of 440 kHz. The digital signal average power is about 15 dB below the host FM.

The FM-1 IBOC system uses 48 spread spectrum data subchannels. The data rate for each channel is 8 kbps, for a total of 384 kbps. The symbol duration is 125 microseconds. For this system 48 subchannels are used. In addition, a 49th subchannel is transmitted as a training signal for multipath equalization.

F. USADR FM-2

The FM-2 stereo audio source coding rates vary from a minimum of 128 kbps to a maximum of 256 kbps on a frame-by-frame basis. The ancillary data is buffered and transmitted at a varying transmission rate and also varied on a frame-by-frame basis.

With the use of frequency shifting techniques, the IBOC digital signal is transmitted orthogonal to the analog host FM. The digital energy extends into the adjacent upper and lower channels.

The FM-2 IBOC system uses 64 data subchannels. The data rate for each channel is 2 kbps. The 8 level amplitude shift key modulation produces a data rate of 3 bits per symbol per subchannel. The symbol duration is 500 microseconds, and the data rate is 384 kbps.

G. USADR AM

The USADR AM audio source coding rate is 96 kbps. An ancillary data stream of 2.4 kbps is included. With forward error correction, overhead brings the modulation data rate to 128 kbps. A 480 ms duration interleaver is used to distribute errors bursts in time. The bandwidth of the composite digital signal, including the analog AM, is 40 kHz.

Appendix 2

Subjective Assessments of Audio Quality of DAR Systems

I. Introduction

This document describes the procedures and results of subjective tests conducted at the Communications Research Centre (CRC), Ottawa, Ontario, Canada, performed to assess the audio quality of digital audio radio (DAR) systems submitted to the Electronic Industries Association's Digital Audio Radio Subcommittee.

A total of nine DAR systems were submitted for testing and are labeled in these results as *a* to *i*. Subjective audio quality was assessed in the absence of any transmission error, thus evaluating the quality of the audio source coding component of each system. One of the nine systems was tested with two different comparison references because the sampling rate for that system was lower than for the other 8 systems, and this report refers to 10 systems noted as *a* to *j*.

II. Subjective Assessment Procedures

A panel of three expert listeners selected final test materials from the initial pool of program segments received from the evaluation subcommittees. This panel selected nine materials, two of which were stressful to each system under test. These are listed in Table 1.

A total of 21 listeners went through the test process for two days each, to complete the 90 rating trials (10 systems x 9 materials). The equipment, listening environment and procedures were the standard ones used in subjective tests at the CRC as described in ITU-R Rec. BS.1116 [1] Statistical evaluations assessed each individual's listening expertise by way of a *t*-test, which showed that no listener who took part in the experiment scored below 2.00. Therefore, they all showed that they were able to discriminate correctly between hidden reference and system versions across all the trials in the experiment.

The actual scale used by the subjects is shown in Figure 1. It is a 5 grade rating scale (1.0 to 5.0) where listeners were instructed to use a single decimal point. In effect, this is a 41 point scale. The subjects were instructed to treat this as a continuous scale but, to facilitate the subjects' orientation, category labels were associated with the scale. Thus, 1.0 to 1.9 is a "very annoying" range; 2.0 to 2.9 is "annoying"; 3.0 to 3.9 is "slightly annoying"; 4.0 to 4.9 is "perceptible but not annoying". Finally, 5.0 is "imperceptible".

The listener's task on a trial is to compare each of two alternative versions of an audio material labeled "B" and "C" with a <u>known</u> Reference version, labeled "A", of the same material. The subject knows that one of the alternatives ("B" or "C") is a "hidden reference", identical to the Reference, and that the other alternative is one that has been processed through a DAR system. The subject does not know which is which, but must decide this through listening. He or she then assigns a grade to both "B" and "C" alternatives, as compared to the known Reference "A", using the 1.0 to 5.0 scale. A is that the alternative the subject has decided is the "hidden reference" must be graded 5.0. And so, *at least one* of the two grades on each trial must be a 5.0

Thus two totally interdependent scores from the listener are recorded on each trial. This deliberate interdependence is handled by subtracting the score given to the true hidden reference from the score given the true processed version (i.e., DSB System minus reference). so that in a graphical plot of outcomes, the data will fall in the same geometric quadrant as they would if the actual 1.0 to 5.0 scores used by the subjects were plotted. Thus the scores are transformed so that the 1.0 to 5.0 range of the original scale becomes, instead, -4.0 to 0.0 in the analysis and presentation of results. These difference grades or "diffgrades" represent the relative differences between the grades given to the hidden reference and the ones given to the DSB system under test.

III. Test Results

For visual clarity, the average quality diffgrades obtained in the experiment are divided between Figures 2(a) and 2(b) rather than being shown within a single graph. Six of them appear in the first figure, four in the second. In addition to the average score among the listeners for each of the audio materials, the overall average diffgrade (the average across all audio materials for each system) is plotted in the "System Averages" column at the right-hand side of these Figures.

Table 2 shows the overall average diffgrade for each audio material and for each system as well as the overall (average) diffgrade for each system in the right-hand column. This table shows all the numbers that are plotted in Figure 2(a) and 2(b). In Table 2, the average diffgrades across all listeners for each audio material occupy a separate row for each DSB system. The average diffgrades are entered to two decimal figures. Systems are arranged by row in alphabetical order using the letters attributed to the ten systems tested -- part of the "double blind" procedures followed throughout the tests..

IV. Overall System Results

The statistical method used to evaluate the present results is the Analysis of Variance (ANOVA) which has been officially recommended in ITU-R Rec. BS.1116 [1]. The experimental design used for these tests permitted the rigorous application of this analytic method. The first item for discussion is the overall average diffgrade for systems. The ANOVA showed that the overall experimental differences among systems in the tests have a very fine resolution of 0.17 of a grade in the transformed diffgrade scale.

For completeness, however, if a reader is interested in evaluating overall differences among audio materials independent of systems (as shown in the averages in the bottom row of Table 2), the critical value provided by the ANOVA is 0.23. This applies to the "without *i* and *j*" averages. Thus, any two of the 9 audio material averages ("without *i* and *j*") across systems must differ by at least 0.23 before they can be considered significantly different on statistical grounds.

The "two" systems (*i* and *j*) rate differences in the references against which subjects compared them. System are actually the same coding system. But they were treated differently in the experiment because of sampling rate differences in the references against which subjects compared them;. System *i* was always compared with 32 kHz sampling rate references, while for system *j*, the references were always sampled at 48 kHz. The ANOVA showed that the overall difference between *i* and *j* were 0.01, well below the 0.17 needed for a conclusion of significant difference.

V. Interaction of Systems with Audio Materials

The ANOVA reveals that the resolution for the interaction of audio materials and systems in this experiment is 0.45 of a grade. This too is a very fine degree of resolution for interactions of this type. When comparing diffgrades between any two systems for any given audio material in Figure 2(a) and 2(b), Table 4 and Figure 3, a numerical difference of 0.45 or greater is required before it can be concluded that those two diffgrades are statistically different from each other rather than being due to chance ($p<0.05$).

VI. Summary

Table 3 shows system identifications in the first column, summarizing the major outcomes using the three criteria developed and used by the ITU-R to evaluate the relative merits of audio coding systems.

First, the overall average diffgrade is shown for each system. This is presented in the second column of the table. Secondly, to summarize the interaction of audio materials by systems and to indicate the size of the variability of each system, the number of times each system fell below a diffgrade of -1.0 for the 9 materials is presented in the third column of the table. To take statistical error into account, the number of times that any system's lower error bar fell "below -1.0" for any material in Figure 3 provided the count shown in this third column. Finally, another ITU-R criterion related to the variability or consistency of each system is shown in the fourth column. This is the number of times that a system could be considered "transparent" for an audio item. The number of times that any system's upper error bar fell above 0.0 in the charts of Figure 3 provided the count shown in this fourth column. Table 3 also shows the systems associated with their letter codes.

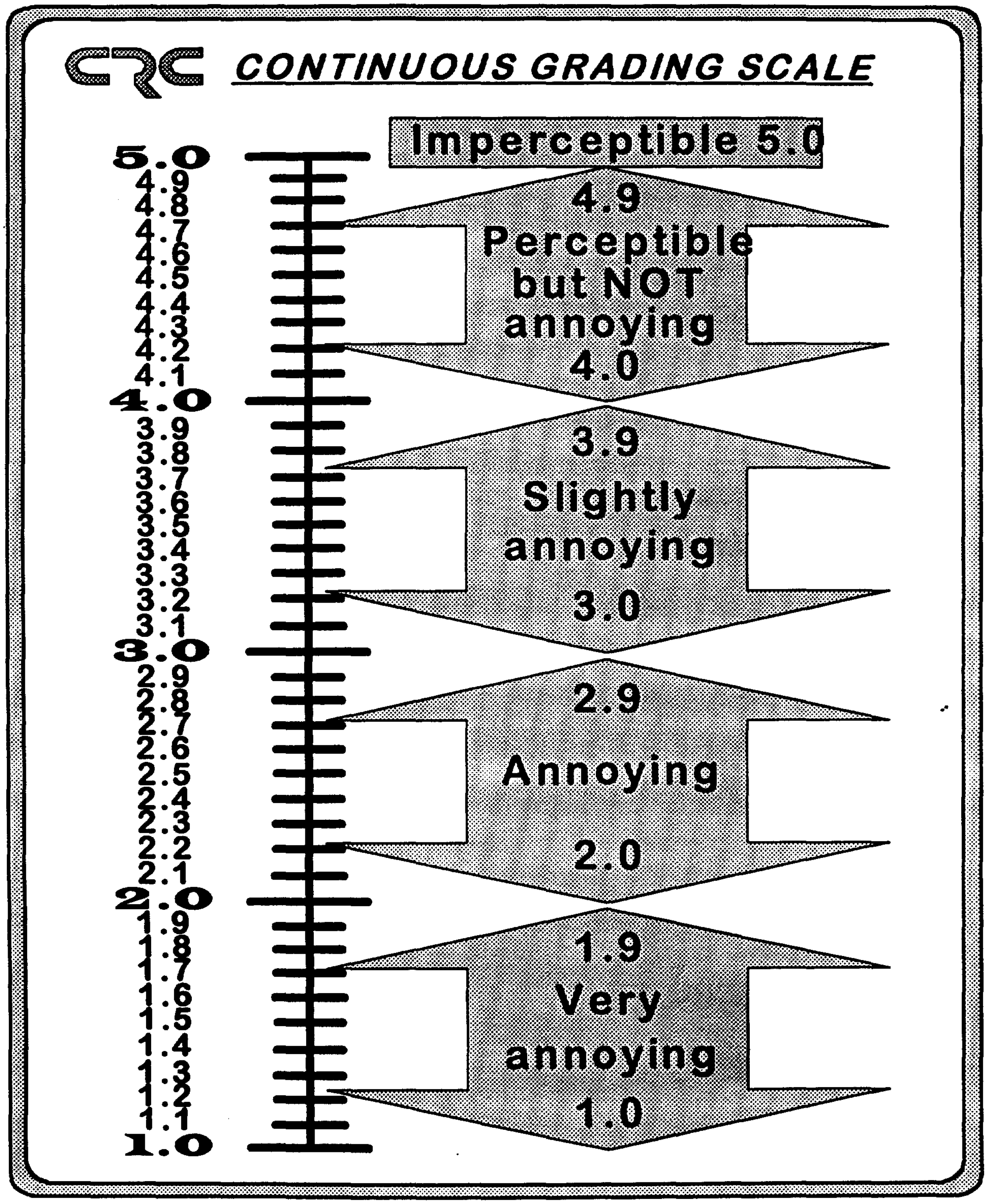


Figure 1 ITU-R continuous 5-grade impairment scale

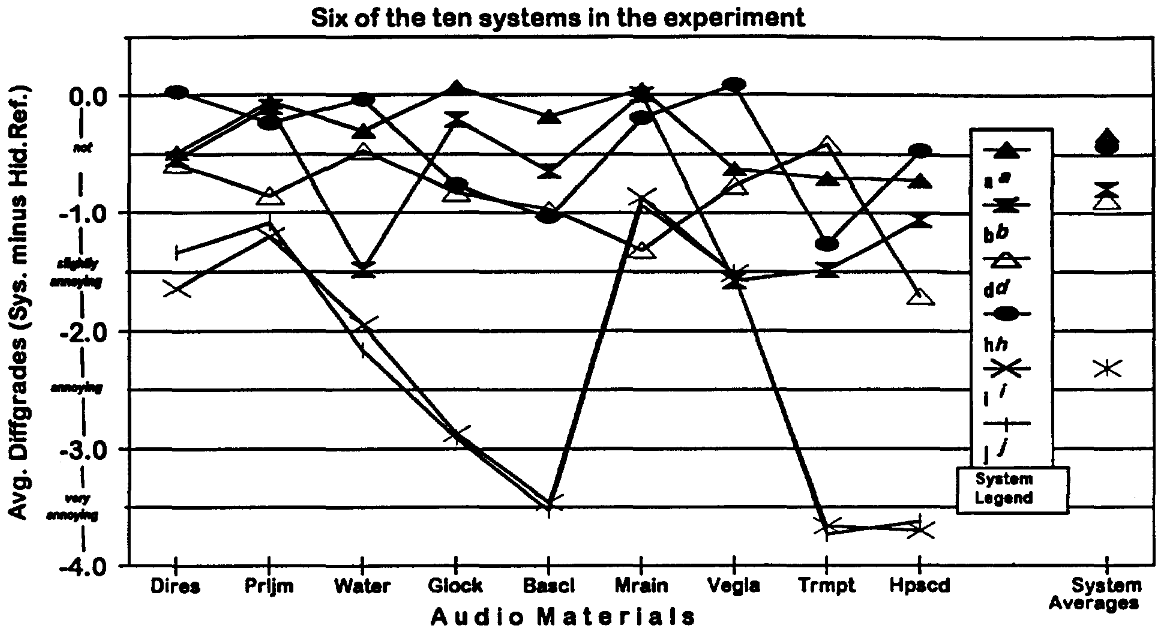


Figure 2(a) Quality test results - systems a,b,d,h,i &j

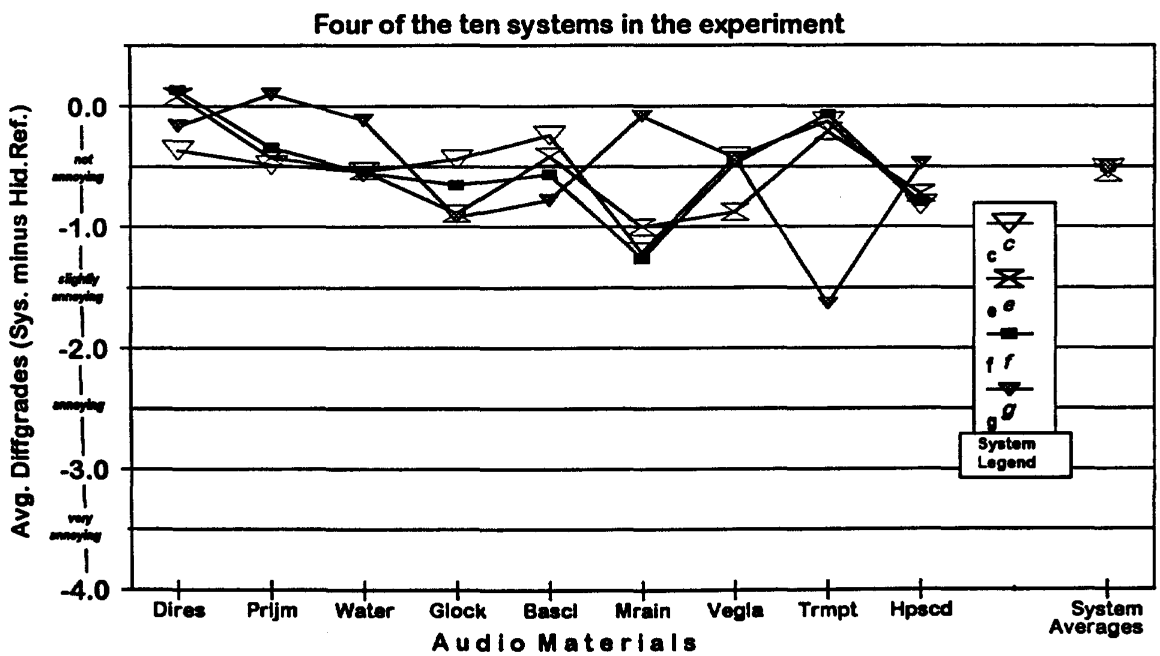
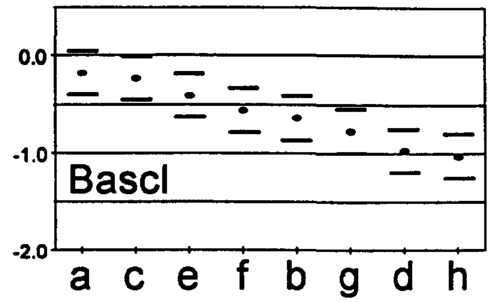
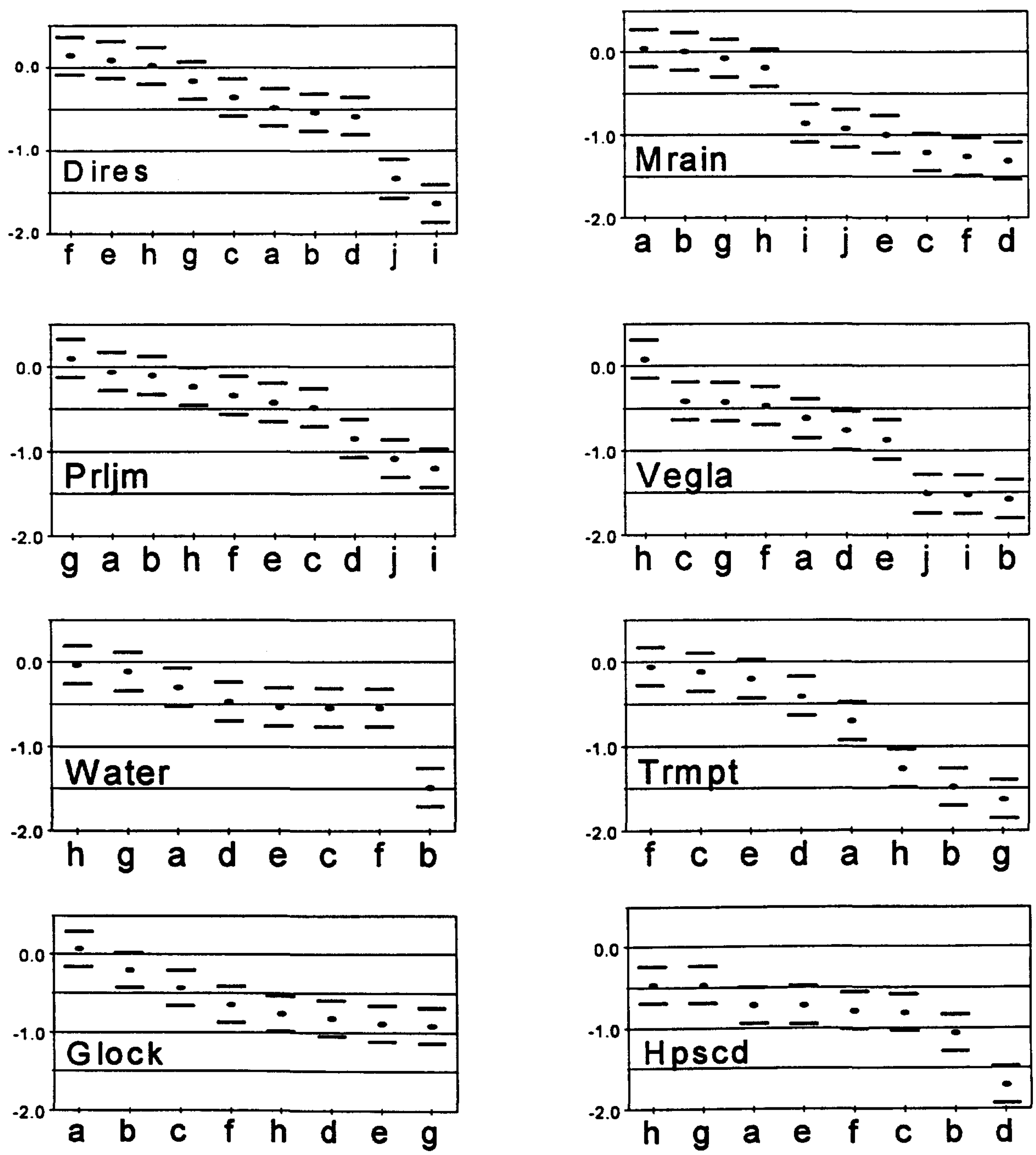


Fig. 2(b) Quality test results - systems *c*, *e*, *f*, and *g*

Fig. 3 System Differences Within Audio Materials
Upper and lower statistical boundaries are shown for the average of each system within each audio material. *Only systems with no horizontal overlaps among their boundaries are statistically different.* Within each chart, systems are ordered along the X-axis by the magnitude of their averages.

The vertical axes start at -2.0 rather than, as in Figs. 1a and b, at -4.0. Systems *i* and *j* are omitted from those charts where their averages fall below -2.0. At those low values, *i* and *j* are significantly different from all the other 8 systems in those audio materials without ambiguity.





Code	Description	Duration	Source
Dires	Dire Straits cut	30 s	Warner Bros. CD 7599-25264-2 (track 6)
Prljm	Pearl Jam cut	30 s	Sony/Epic CD ZK53136 (track 3) with processing[1]
Water	Sounds of water	30 s	Roland Dimensional Space Processor Demo. CD
Glock	Glockenspiel	16 s	EBU SQAM CD (track 35/Index 1)
Bascl	Bass Clarinet arpeggio	30 s	EBU SQAM CD (track 17/Index 1) with processing[1]
Mrain	Music and rain	11 s	AT&T mix
Vegla	Susan Vega with glass	11 s	AT&T mix
Trmpt	Muted trumpet	9 s	Original DAT recording, University of Miami
Hpscd	Harpsichord arpeggio	12 s	EBU SQAM CD (track 40/Index 1)

[1] Processing chain used: Aphex Compellor Model 300 (set for leveling only)
Dolby Spectral Processor Model 740
Aphex Dominator II Model 720

Table 1 List of audio test materials used in the quality tests

The data for a single system are shown throughout each row.

System	Dires	Prljm	Water	Glock	Bascl	Mrain	Vegla	Trmpt	Hpscd		Overall Averages
a	-0.49	-0.06	-0.30	0.07	-0.18	0.04	-0.62	-0.70	-0.72	*a*	-0.33
b	-0.54	-0.10	-1.49	-0.21	-0.64	0.00	-1.58	-1.49	-1.07	*b*	-0.79
c	-0.36	-0.49	-0.54	-0.44	-0.24	-1.21	-0.42	-0.12	-0.82	*c*	-0.52
d	-0.59	-0.85	-0.47	-0.82	-0.97	-1.31	-0.77	-0.41	-1.70	*d*	-0.88
e	0.09	-0.43	-0.53	-0.89	-0.41	-1.00	-0.88	-0.20	-0.72	*e*	-0.55
f	0.14	-0.34	-0.55	-0.65	-0.57	-1.26	-0.47	-0.06	-0.80	*f*	-0.51
g	-0.16	0.10	-0.11	-0.92	-0.78	-0.08	-0.43	-1.63	-0.48	*g*	-0.50
h	0.02	-0.24	-0.04	-0.77	-1.04	-0.20	0.08	-1.27	-0.47	*h*	-0.43
i	-1.64	-1.20	-1.95	-2.87	-3.46	-0.86	-1.52	-3.66	-3.70	*i*	-2.32
j	-1.34	-1.09	-2.16	-2.91	-3.52	-0.93	-1.51	-3.73	-3.62	*j*	-2.31
Audio Material Averages	-0.49	-0.47	-0.81	-1.04	-1.18	-0.68	-0.81	-1.33	-1.41		-0.91
Averages Without *i* and *j*	-0.24	-0.30	-0.50	-0.58	-0.60	-0.63	-0.64	-0.74	-0.85		-0.56

System *i* received a grade of -1.95 for Water. In view of the statistical error (0.45 of a grade), *i* was omitted from Water in Fig. 2.3 on the next page, along with other instances of *i* and *j* in materials where either of these two systems obtained a diffgrade lower than -2.00. (No systems other than *i* and *j* received any diffgrades below -2.00.)

Table 2: Average Difference Grades for each of the 9 Audio Materials (columns) by each of the 10 Systems

System Designation	Overall Average Diffgrade	Number of transparent materials	Number of materials below -1.0
A - Eureka 147, MUSICAM @ 224 kbps	-0.33	4	0
B - Eureka 147, MUSICAM @ 192 kbps	-0.79	3	4
C - AT&T/Lucent, PAC @ 160 kbps	-0.52	2	1
D - AT&T/Amati, DSB PAC @ 160 kbps	-0.88	5	0
E - AT&T/Amati, LSB PAC @ 160 kbps	-0.55	3	2
F - VOA/JPL, PAC @ 160 kbps	-0.51	2	2
G - USADR FM-2, MUSICAM @ 256 kbps	-0.50	2	4
H - USADR FM-1, MUSICAM @ 256 kbps	-0.43	2	4
I - USADR AM, MUSICAM @ 96 kbps (32 kHz reference)	-2.32	0	9
J - USADR AM, MUSICAM @ 96 kbps (48 kHz reference)	-2.31	0	9

Table 3
Summary of Audio Quality Tests

Appendix 3

Digital Sound Broadcasting Impairment Test Results

Introduction

This document is intended to focus only on the digital impairment tests for all seven systems. Complete laboratory test results for all seven systems are available from EIA.

Up to three audio test segments that originated from the EBU SQAM disc, glockenspiel, soprano, and clarinet were used for transmission impairment tests.

The desired signal receiver input level for the impairment tests was -62 dBm for the systems in the FM band systems and -60 dBm for the L and S band.

Gaussian Noise, Co-Channel, and Multipath and Noise Tests

For the noise test filtered gaussian noise was added to the signal and the noise increased until the threshold of audibility was heard by the laboratory specialists. The Threshold Of Audibility (TOA) is the point where the interference is just perceptible. From the TOA the noise was further increased until the point of failure was heard. Point Of Failure (POF) is the point where the signal completely fails or the interference is very annoying. A remotely controlled 0.25 dB steps attenuator was used to find the TOA and POF. Digital audio tapes were made with the added noise level ranging from below TOA to a level above POF. These recordings were used for further subjective assessment at the Communications Research Centre. Laboratory type average power meters were used to measure signal power.

Table #1 shows the results of the noise tests with the three audio segments. To compensate for the differing digital bandwidths (0.2 MHz to 1.5 MHz), the performance for added noise was calculated using C_o/N_o. The TOA/POF noise spreads varied 4.2 dB from shortest to the longest. The AT&T IBAC system had a 0.8 dB spread and the USADR FM-1 IBOC had a 5 dB spread.

Co-Channel

Each proponent supplied a second system transmitter or a system simulator for the co-channel tests. The co-channel signal was increased in 0.25 dB steps until the TOA and POF were heard by the laboratory specialists. The results of the tests are in desired/undesired (D/U) signal ratios. Digital audio tapes were recorded with the co-channel ranging from just below TOA to above POF for further subjective assessment at the CRC.

Table #2 shows the results of the co-channel tests. The TOA/POF spreads for co-channel were slightly higher than those for noise.

Multipath and Noise Tests

The simulated multipath and noise tests were conducted twelve times, each with a different multipath scenario: urban slow, urban fast, rural fast, and terrain obstructed, using three audio segments for each scenario. The multipath parameters were specified by the channel characterization sub-group. Digital recordings were made for further subjective assessment at the CRC.

Table #3 shows the laboratory test results with three audio test materials and the four multipath scenarios. If impairments were heard without noise added, the signal audio was rated by the lab experts. For those multipath tests where no impairment was heard, noise was added in 0.5 dB steps until the TOA and POF were found. The numerical results of the tests are in Desired/Undesired (D/U) signal ratios. These tests were recorded and sent to the CRC for further assessment.

If multipath impairments were heard by the laboratory experts without noise added, Expert Observation and Commentary (EO&C) tests were conducted by the transmission laboratory experts. The scale for the EO&C tests is shown in the table.

Co-Channel, First and Second Adjacent Without Multipath

These tests measured the Digital to Digital interference to co-channel, first adjacent, and the second adjacent. The adjacent channel tests were conducted on both the lower and upper channels. The undesired signal was increased in 0.5 dB steps until the TOA and POF were heard by the laboratory specialists. The EBU SQAM disc glockenspiel was used for the test audio. For the Inband-On/Channel (IBOC) systems, the composite signal was used. The EO&C tests were conducted by the transmission laboratory specialists. The D/U at the TOA and POF is reported for each system. Table #4 shows the results of these tests.

Co-Channel, First, and Second Adjacent Channels with Multipath

These tests measured the Digital to Digital interference to co-channel, lower first adjacent, and lower second adjacent. The undesired signal was increased in 0.5 dB steps until the TOA and POF were heard by the laboratory specialists. If interference was heard without undesired signal added, no additional assessments were conducted. Glockenspiel was used for the test audio. The D/U at the TOA and POF is reported.

Tables #5, #6, and #7 show the results of the interference tests with multipath. The assessments were completed by the specialist at the transmission laboratory.

Re-Acquisition

Noise was added to the signal in 0.25 dB steps until POF. At POF the attenuator setting was recorded. The DAR transmitter was then disconnected from the receiver for at least 30 seconds to assure loss of lock. The signal was then reconnected to the DAR receiver and acquisition time recorded. Acquisition is the reproduction of usable music. Mozart track 67 of the EBU SQAM disc was used. The test was conducted three times with the noise set at 2 dB, 4 dB, and 6 dB below POF. At each noise level the test was conducted five times, and the results were averaged. The results of the re-acquisition tests with simulated multipath are not included in this document.

Table 8 shows the average results of the five tests in seconds. POF-2, POF-4, and POF-6 represent the signal levels below POF. The assessments were completed by the specialist at the transmission laboratory.

GAUSSIAN NOISE

PROPONENT	GLOCKENSPIEL		SOPRANO		CLARINET	
	C_o/N_o	C_o/N_o	C_o/N_o	C_o/N_o	C_o/N_o	C_o/N_o
	dB	dB	dB	dB	dB	dB
	TOA	POF	TOA	POF	POF	TOA
A E-147 224 Kb/s	8.48	5.98	8.23	6.23	8.98	6.48
B E-147 193 Kb/s	8.46	5.96	8.71	6.21	8.96	6.46
C AT&T	11.36	10.61	11.11	10.36	11.11	10.36
D LSB AT&T/AMATI	18.85	16.85	17.60	16.35	18.10	16.60
E DSB AT&T/AMATI	10.76	9.51	10.51	9.51	10.76	9.51
F JPL VOA	3.26	2.26	3.26	2.26	3.26	2.51
G FM2 USADR	25.10	21.60	25.10	21.35	26.35	22.35
H FM1 USADR	10.51	8.51	10.01	8.51	10.51	8.51
I AM USADR	19.64	17.14	19.64	17.64	19.89	17.89
K DSB AT&T/AMATI	10.29	8.79	10.04	8.79	10.04	8.79
L FM1 USADR	11.33	6.33	10.83	6.83	11.08	6.58

Table 1